the PBX without delivering the performance provided by dedicated LANs.

A typical office today thus uses two separate and independent networks: a PC LAN to distribute computer data, and a PBX to provide telephone services. The hardware infrastructure of the two networks is independent and separate. Each network requires its own dedicated physical connection medium such as coaxial cable, twisted pair wiring, etc. Traditionally, PBX switching equipment, terminal equipment, control computer resources and in-house wiring are separate devices, not shared or leveraged by the two networks.

The term computer telephone integration (CTI) describes any system which employs a computer to enhance or control telephony. This is implemented by interfacing PBXs and computers, bringing caller information to the computer so database lookup and screen pops to the called agent are possible. Other implementations utilize separate servers with new buses to add voice processing capability. Recently, CTI developers have developed equipment which, when added to a standard PC, allows the functions of a PBX to be implemented. The same PC which operates on the LAN may now also be used to implement the PBX.

Despite the integration afforded by CTI, artifacts of the different development of PBXs and LANs remain. Although the PBX and LAN may be implemented by a standard PC, and may even physically reside within the same device, the two networks remain separate and independent systems. The LAN continues to use its own data transport protocol and physical connection media to each device on the network. The PBX uses its telephony signaling scheme, switching equipment, and separate dedicated physical connection media to transmit voice data.

More recently, ATM (asynchronous transfer mode) networks have been envisioned to integrate digital data

with multimedia voice and video onto a single high speed line or "pipe". ATM packages and transmits digital data in small 53 byte fixed-length messages or cells while providing high bandwidths of 25 Mbps and higher. Although ATM networks were envisaged to provide transport of data, voice and video, little has been done to facilitate the transmission of real-time, low latency voice traffic on ATM local area networks. ATM voice transmission efforts to date have primarily been focused on higher-capacity wide area networks, campus backbones and longer haul transmission networks.

The ATM forum has developed ATM standards for local area networks. A great strength of ATM is the ability of the network to assign an appropriate quality of service (QoS) class to a particular transmission. ATM networks can guarantee that strict requirements on available bandwidth and minimal delay can be guaranteed for those connections requiring predictable service. This makes reliable voice transmission possible over an ATM network. Although the bandwidth requirements for voice are easily met by other local area networking technologies, ATM can today provide the predictable quality of service required for real-time bi-directional communication.

## SUMMARY OF THE INVENTION

The present invention uses CTI to implement a distributed private branch telephone exchange (PBX) over a local area computer network or LAN. The system leverages the power of desktop PC's through graphical user interfaces (GUIs) and standard interfaces such as Object Linking and Embedding (OLE) to simplify and extend conventional telephony. The LAN telephone system includes a unique multi-port station module in each desktop client computer that provides both the network data interface and an interface to a standard telephone set. Quality voice transmission is achieved by the use of

real-time voice streaming, which directly converts digitized voice to cells ready for transmission over the asynchronous network or for local storage on the computer hard drive for later playback.

A different network module (or modules in larger systems) plugs into the network server. This PSTN module 20 interfaces the LAN telephone system to outside trunk lines provided by the local telephone company. It combines telephone trunk interfaces with digital signal processing for caller ID, DTMF and call progress detection, and real-time voice streaming to facilitate transmission of voice within the LAN.

The desktop client computers are linked to each other and the server through an ATM switch, which transmits network traffic using the conventional ATM protocols as defined in ATM Forum standards. Using the unique adapter modules described in this patent allows the network to support not only conventional ATM traffic, but also the transport of high quality voice transmission, and the conversion of voice information from analog or digital signals to ATM cells and back.

Another component of the system is the telephone hub, which allow the use of telephones not associated with computers; for example, telephones on a production floor, or in conference rooms. In the preferred embodiment, this device connects the hub to the network via a LAN connection, and allows connection to eight or more telephones.

The system includes software that uses this unique voice-enabled LAN to implement a distributed PBX that controls the initiation and termination of telephone calls between telephone handsets attached to client PC's, to telephone hubs, and via outside trunk connections to the PSTN. This PBX differs from previous implementations in that a standard ATM LAN has been used to replace the usual backplane connection between trunk and station line interfaces, and that voice transmissions are carried over

the came set of wires as LAN data. Conventional PBX signaling between trunk and station, or station and station, has been translated into network messages that convey information relating to real-time telephony events on the network, or instructions to the network adapters to generate the appropriate signals and behavior to support normal voice communication, or instructions to connect voice media streams using standard ATM connection and signaling protocols.

The control software of the PBX runs on one computer on the network, usually the server, (or servers in large systems), and includes a network telephony services provider. Telephony applications, including voice mail, auto attendant, CTI applications, a client Telephone Assistant graphical user interface (GUI), configuration and administration GUIs, and an operator console GUI are implemented on the network of server and client computers.

## BRIEF DESCRIPTION OF THE DRAWINGS

Fig. 1 is diagram of an embodiment of the telephone network of the present invention.

Fig. 2 shows a diagram of the multi-port PSTN module of the telephone network of Fig. 1.

Fig. 3 shows a diagram of the multi-port station module of the telephone network of Fig. 1.

Fig. 4 shows a block diagram of the telephony hub with 8 telephones.

Fig. 5 illustrates real-time voice streaming performed by the telephone network of Fig. 1.

Fig. 6 shows a high level diagram of the control logic of the multi-port PSTN module of Fig. 2.

Fig. 7 shows the server software architecture of the telephone network of Fig. 1.

Fig. 8 shows the client software architecture of the telephone network of Fig. 1.